Claims

What is claimed is:

1. A method of converting a first digital audio signal at a first sampling rate to a second digital audio signal at any of a plurality of sampling rates that are higher than said first sampling rate, said method comprising:

up-sampling said first digital audio signal by an up-sampling factor to generate an up-sampled audio signal;

low-pass filtering said up-sampled audio signal using a fixed set of filter coefficients to generate a filtered audio signal; and

down-sampling said filtered audio signal by any of a plurality of down-sampling factors corresponding to said plurality of sampling rates, wherein each of said plurality of down-sampling factors is lower than said up-sampling factor, to generate said second digital audio signal at any of said plurality of audio sampling rates.

2. The method of claim 1 wherein said down-sampling includes performing linear interpolation between filtered samples of said filtered audio signal to generate

interpolated output samples of said second digital audio signal.

3. The method of claim 1 wherein said up-sampling factor is an integer.

4. The method of claim 1 wherein said plurality of down-sampling factors are non-integers.

5. The method of claim 1 wherein a cut-off frequency, corresponding to a frequency response of said fixed set of filter coefficients, is no greater than a frequency corresponding to π divided by said up-sampling factor.

6. The method of claim 2 wherein said performing linear interpolation is done in response to an accumulated linear interpolation ratio that is based on a current down-sampling factor of said plurality of down-sampling factors and a current phase of said second digital audio signal.

7. The method of claim 1 wherein said up-sampling, said low-pass filtering, and said down-sampling, to generate at least one interpolated output sample of said second digital audio signal, comprises:

storing said fixed set of filter coefficients as a plurality of phased subsets of filter coefficients;

applying samples of said first digital audio signal to at least two phased subsets of said plurality of phased subsets of filter coefficients to generate at least two filtered samples of said filtered audio signal; and

linear interpolating between said at least two filtered samples to generate said at least one interpolated output sample of said second digital audio signal.

8. The method of claim 7 wherein a total number of said plurality of phased subsets is equal to said up-sampling factor.

9. The method of claim 7 wherein said at least two phased subsets are selected from said plurality of phased subsets to generate said at least two filtered samples based on a current down-sampling factor of said plurality of down-sampling factors and a current phase of said at least one interpolated output sample.

10. The method of claim 7 wherein said linear interpolation is done in response to an accumulated linear interpolation ratio that is based on a current down-sampling factor of said plurality of down-sampling factors and a current phase of said at least one interpolated output sample.

11. Apparatus for converting a first digital audio signal at a first sampling rate to a second digital audio signal at any of a plurality of sampling rates that are higher than said first sampling rate, said apparatus comprising:

a memory storing a fixed set of digital filter coefficients;

a low-pass filter applying said fixed set of digital filter coefficients to said first digital audio signal to generate a filtered audio signal; and

a linear interpolator generating an accumulated linear interpolation ratio and applying said accumulated linear interpolation ratio to said filtered audio signal to generate said second digital audio signal at any of said plurality of sampling rates wherein said plurality of sampling rates are higher than said first sampling rate.

12. The apparatus of claim 11 wherein said fixed set of digital filter coefficients are arranged in said memory as a plurality of phased subsets of said fixed set of digital filter coefficients.

13. The apparatus of claim 12 wherein a total number of said plurality of phased subsets is equal to an effective up-sampling factor.

14. The apparatus of claim 13 wherein said effective up-sampling factor is an integer.

15. The apparatus of claim 11 wherein said accumulated linear interpolation ratio is based on a current effective down-sampling factor and a current phase of said second digital audio signal.

16. The apparatus of claim 15 wherein said effective down-sampling factor is a non-integer.

17. The apparatus of claim 11 wherein said memory is a ROM.

18. The apparatus of claim 12 wherein a current interpolated output sample of said second digital audio signal is generated from at least two current filtered samples of said filtered audio signal.

19. The apparatus of claim 18 wherein said at least two current filtered samples of said filtered audio signal are generated by applying at least two phased subsets of said plurality of phased subsets.

20. The apparatus of claim 19 wherein said at least two phased subsets are selected from said plurality of phased subsets to generate said at least two current filtered samples based on a current down-sampling factor and a current phase of said second digital audio signal.